FORM-PTO-1390 (Rev. 12-29-99)	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTORNEY'S DOCKET NUMBER
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371		032287-023
		U.S APPLICATION NO. (If known, see 37 C.F.R 1.5) UNASSIGNED 09/869367

INTERNATIONAL APPLICATION NO. PCT/AT99/00278	INTERNATIONAL FILING DATE 17 November 1999	PRIORITY DATE CLAIMED 30 December 1998

TITLE OF INVENTION
TRANSMISSION SYSTEM WITH ECHO CANCELLATION

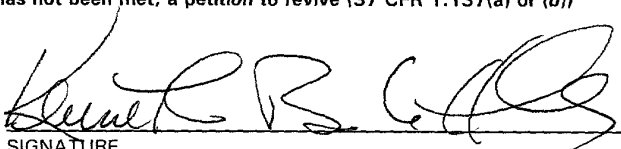APPLICANT(S) FOR DO/EO/US
Gerhard ZIMMERMANN

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.

2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.

3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and the PCT Articles 22 and 39(1).

4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.

5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2))

    a. ☒ is transmitted herewith (required only if not transmitted by the International Bureau).

    b. ☒ has been transmitted by the International Bureau.

    c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US)

6. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).

7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3))

    a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).

    b. ☐ have been transmitted by the International Bureau.

    c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.

    d. ☒ have not been made and will not be made.

8. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).

9. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)).

10. ☒ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

Items 11. to 16. below concern other document(s) or information included:

11. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.

12. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.

13. ☒ A FIRST preliminary amendment.

    ☐ A SECOND or SUBSEQUENT preliminary amendment.

14. ☐ A substitute specification.

15. ☐ A change of power of attorney and/or address letter.

16. ☒ Other items or information:

    International Preliminary Examination Report, Unexecuted Declaration

U.S. APPLICATION NO. (If known, see 37 C.F.R. 1.50) UNASSIGNED 09/869367	INTERNATIONAL APPLICATION NO. PCT/AT99/00278	ATTORNEY'S DOCKET NUMBER 032287-023

17. ☒ The following fees are submitted:	CALCULATIONS	PTO USE ONLY

Basic National Fee (37 CFR 1.492(a)(1)-(5)):

Neither international preliminary examination fee (37 CFR 1.482)
nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO
and International Search Report not prepared by the EPO or JPO $1,000.00 (960)

International preliminary examination fee (37 CFR 1.482) not paid to
USPTO but International Search Report prepared by the EPO or JPO $860.00 (970)

International preliminary examination fee (37 CFR 1.482) not paid to USPTO
but international search fee (37 CFR 1.445(a)(2)) paid to USPTO $710.00 (958)

International preliminary examination fee paid to USPTO (37 CFR 1.482)
but all claims did not satisfy provisions of PCT Article 33(1)-(4) $690.00 (956)

International preliminary examination fee paid to USPTO (37 CFR 1.482)
and all claims satisfied provisions of PCT Article 33(1)-(4) $100.00 (962)

	CALCULATIONS	PTO USE ONLY
ENTER APPROPRIATE BASIC FEE AMOUNT =	$ 860.00	
Surcharge of **$130.00 (154)** for furnishing the oath or declaration later than 20 ☐ 30 ☐ months from the earliest claimed priority date (37 CFR 1.492(e)).	$ -0-	

Claims	Number Filed	Number Extra	Rate		
Total Claims	13 -20 =	-0-	X$18.00 (966)	$ -0-	
Independent Claims	2 -3 =	-0-	X$80.00 (964)	$ -0-	
Multiple dependent claim(s) (if applicable)			+$270.00 (968)	$ -0-	

TOTAL OF ABOVE CALCULATIONS =	$	
Reduction for 1/2 for filing by small entity, if applicable (see below).	$ -0-	-
SUBTOTAL =	$	
Processing fee of **$130.00 (156)** for furnishing the English translation later than 20 ☐ 30 ☐ months from the earliest claimed priority date (37 CFR 1.492(f)). +	$ -0-	
TOTAL NATIONAL FEE =	$ 860.00	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31). **$40.00 (581)** per property +	$ -0-	
TOTAL FEES ENCLOSED =	$ 860.00	
	Amount to be: refunded	$
	charged	$

a. ☐   Small entity status is hereby claimed.

b. ☒   A check in the amount of $ 860.00 to cover the above fees is enclosed.

c. ☐   Please charge my Deposit Account No. 02-4800 in the amount of $_____ to cover the above fees. A duplicate copy of this sheet is enclosed.

d. ☒   The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. 02-4800. A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

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Kenneth B. Leffler
NAME

36,075
REGISTRATION NUMBER

(01/01

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Patent Application of	)
	)
Gerhard ZIMMERMANN	) Group Art Unit: UNASSIGNED
	)
Application No.: UNASSIGNED	) Examiner: UNASSIGNED
	)
Filed: June 28, 2001	)
	)
For: TRANSMISSION SYSTEM WITH ECHO	)
CANCELLATION	)
	)

PRELIMINARY AMENDMENT

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

Prior to examination, please amend the above-identified application as follows:

IN THE CLAIMS:

Please replace claims 3 and 7 as follows:

3. (Amended) Transmission system according to claim 1, with an echo cancellation unit comprising a delay line consisting of several delay elements, wherein the first delay element of the delay line has a relatively high memory delay time that is substantially equal to the minimum overall propagation time of the voice data signals in both directions of the data network.

7. (Amended) Method according to claim 5, wherein when one or several data packets have gotten lost, the respective preceding voice data packet is repeated.

Please add the following new claim 13:

13. (NEW) Method according to claim 4, wherein when one or several data packets have gotten lost, the respective preceding voice data packet is repeated.

REMARKS

The above changes to the claims have been made to delete multiple dependency of the claims, to round out the scope of patent protection being sought, and generally to place the claims in better condition for examination on the merits.

Respectfully submitted,

BURNS, DOANE, SWECKER & MATHIS, L.L.P.

By: _____
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Date: June 28, 2001

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Attachment to Amendment dated June 28, 2001

Marked-up claims 3 and 7

3. (Amended) Transmission system according to claim 1, [or 2] with an echo cancellation unit comprising a delay line consisting of several delay elements, wherein the first delay element [(20)] of the delay line has a relatively high memory delay time that is substantially equal to the minimum overall propagation time of the voice data signals in both directions of the data network [(1)].

7. (Amended) Method according to [one of the claims 4 or 5] claim 5, wherein[,] when one or several data packets have [got] gotten lost, the respective preceding voice data packet is repeated.

TRANSMISSION SYSTEM WITH ECHO CANCELLATION

The invention relates to a transmission system for transmitting speech information within at least one data transmission network, such as LAN, Intranet, Internet, connecting several subscribers, in which transmission is carried out by means of data packets on the basis of at least one shared protocol, e.g. Internet protocol, and each subscriber is connected to the network via a voice data transmission unit, possibly an analog-to-digital converter and a digital-to-analog converter, comprising a transceiver unit and a voice data conversion unit as well as a speaker and an earpiece unit connected to said voice data transmission unit, the speaker unit or the inserted analog-to-digital converter being connected to the transmission unit via an addition input of an echo cancellation unit and the output thereof, and a subtraction input of the echo cancellation unit being connected to the earpiece unit or to the inserted digital-to-analog converter.

In networks such as, e.g. LAN, Intranet, Internet or the like which foot on a standardized protocol, e.g. the Internet protocol, each network subscriber is assigned an address from which data are sent to other subscribers or at which data sent by other subscribers can be received. Since speech may also be transmitted in the form of digital data, such data transmission networks may also be used for transmitting speech. Within the scope of the invention, the type of the network and of the transfer protocol are not limited to the above mentioned known denominations, the invention may also be used for any connected networks that may be interconnected through gateways at certain points.

In heretofore known speech transmission systems of the type mentioned herein above, speech transmission is carried out in defined packets that are sent from the transmitter station to a receiving subscriber which receives them. For this purpose, the packets are composed in consecutive sequence of the digitized sampling values of the speech signal to be transmitted and can be coded accordingly for transmission on the transmitter side and decoded upon reception on the side of the receiver. The received sampling values are again combined into a speech signal, speech signal meaning all the acoustic vibrations lying in the audible range. Unlike transmission over telephone lines, the peculiarity of speech transmission in data transmission networks consists in that the transmission bandwidth is not always guaranteed by the provider. Depending upon the presence of free lines, constant alterations of the path provided over which the data packets are transmitted likewise occur at different nodes. On account of the differing transit times or the differing data paths, data packets that have been sent later may arrive earlier at the receiver's than data packets sent earlier. Due to such delays, it may happen that discrete data packets do not arrive at the right time at the receiver's, thus not being available in time. Furthermore, the data packets may get lost on account of transmission errors.

In greater networks such as e.g. Internet, the variations in transmission time delays are clearly noticeable and depend on the respective load placed on said network at this very moment by the different subscribers. Besides changes due to the time of the day, certain occurrences may bring about retardations that are in no way foreseeable.

A system for switching and transmitting speech signals over a packet switching net, more specifically over the Internet, and a conventional telephone network is described in WO-A-97/14238 for example. This system permits a telephone connection from a telephone terminal to a computer connected to the internet.

In addition to the delay occurring in the transmission network, echo signals also disturb speech transmission. The acoustic coupling between a talking device, e.g. a microphone, and a hearing device, e.g. a loudspeaker, a phone receiver or the like in a handsfree equipment, in an earpiece or the like, or the electric coupling caused by a hybrid circuit of an analog two-wire phone line can cause such echo signals to form.

Echo signals occur more specifically when the acoustic signals generated by a speaker return, more or less delayed, to his ear. When the time delay of the sound is relatively short, e.g. less than 1 ms, the arising echo does not cause any disturbances since the natural acoustic coupling between mouth and ear or sound reflections in the surroundings via mouth, wall and ears also occasion delays of the same dimension to happen, the human ear being permanently exposed to such delays so that they are no longer consciously perceived by men.

With longer echo delay times that are generated by delays in the transmission path as they occur in speech transmission in packet oriented data networks and which may amount to up to several 100 ms, the echo, depending on its loudness, disturbs or makes communication almost impossible.

Echos can be suppressed by appropriate subtraction of the delayed receiver signal from the signal to be transmitted. This may be performed either in the analog form by means of summing circuits or in the digital form by preferably using a Digital Signal Processor (DSP). The invention is suited to both analog and digital echo cancellation.

Since expenditure and complexity of an echo cancellation unit is proportional to the maximum echo delay time, the attempt is being made to locally suppress the echos on either side of the network, that is on both the receiver and the transmitter side when transmitting speech over the data network, thus to prevent echos from being transmitted over the data network.

In case however, the echo cancellation system should fail to work on the side of one subscriber, or should it not have been implemented owing to the cost it would have involved, the echo created there will be transmitted unhindered to the subscriber connected with the first one and will disturb there the

course of the conversation.

The conventional echo cancellation units that can overcome such echos are mainly designed for transmissions over the telephone network telephony system and therefore for fixed, relatively short delay times. For speech transmission over a packet oriented data network with delay times ranging from 500 ms to 1500 ms though, these known units are only efficient to a limited extent.

Due to the specific properties of a packet oriented speech transmission, a conventional echo cancellation unit may even cause the echo behavior to worsen. If, on account of the permanently changing network properties for example, the echo signal is shifted relative to its original time position, both the echo and the delayed and inverted signal produced for echo cancellation are suddenly to be found in the receiver channel and are no longer canceling out. This condition is kept up until the echo cancellation unit has adjusted anew.

If on the contrary, individual data packets get lost during transmission, the inverse signal suddenly becomes audible because it does not meet with the lost signal to be erased.

It is therefore the object of the invention to indicate a transmission system of the type mentioned herein above that makes it possible, in a technically easy and reliable manner, to erase echos for high delay times, more specifically in packet oriented data transmissions.

This is achieved according to the invention by providing an additional echo cancellation unit with an addition and a subtraction input, the output thereof being connected to the earpiece unit or to the inserted digital-to-analog converter and wherein the subtraction input is connected to the transmitter unit of the voice data transmission unit and the addition input is connected to the receiver unit of the voice data transmission unit.

As a result thereof, the echo generated by the subscriber located on the respective other side of the data transmission network is eliminated by subtracting from the returned echo signal the voice signal(s) or speech information originally produced in the speaker unit and delayed by at least the network delay time. As described herein above, subtraction may be carried out either in the analog or digital form.

In another embodiment of the invention, there may be provided that the additional echo cancellation unit has a control input for controlling a memory delay time of the voice signal or of the speech information that corresponds to the minimum delay time of the network, the control input being connected to the output of a control unit connected to the voice data transmission unit.

The control unit allows to determine the instant network delay time or to detect lost data packets and to suppress an accordingly delayed and inverted transmitter signal in order to prevent the inverted signal formed in replacement of the lacking data packet from reaching the earpiece unit.

According to another embodiment of the invention, there may be provided in a transmission system with an echo cancellation unit comprising a delay line consisting of several delay elements that the first delay element of the delay line has a relatively high memory delay time that is substantially equal to the minimum overall delay time of the voice data signals in both directions of the data network.

Although echo signals may even occur in a delay scope of max. 64 ms, the overall delay time for the data network is ten- to twenty-fold. The expenditure for the delay elements and for the calculation of the coefficients related thereto increases proportionally to the maximum echo delay time. Instead of providing for a conventional delay line with an evenly distributed number of delay elements with tappings for weighting the delayed signal with coefficients k arranged therein between, the first delay element fitted with a relatively high memory delay time makes a delay time possible that corresponds to the minimum overall delay time of the voice signals in both directions. Transmission cannot be carried out with less delay time, so that the number of delay elements with tappings corresponding to said delay time can be combined into this first delay element. The echo cancellation unit then operates with a basic delay of, e.g. 600 ms and with a variable delay range of, e.g. 600 ms to 800 ms. On account of the overall delay time within the data transmission network, no echo signals can form within the range of the basic delay so that the corresponding tappings and coefficients k may be saved. The advantage now lies in the fact that, as a result thereof, the echo cancellation unit spends much less computing power and memory since both less coefficients k need to be computed and less multiplications must be carried out for computing the inverse delayed signal.

According to another embodiment of the invention, the memory delay time of the first delay element of the delay line can be controlled through the voice data transmission unit, preferably by interpreting the time information of the real-time protocol. This makes it possible to permanently adjust the basic delay time of the transmission network, so that the number of the required delay elements may be kept accordingly low.

Furthermore, the invention relates to a method of transmitting speech information within at least one data transmission network, such as LAN, Intranet, Internet, connecting several subscribers, in which transmission is carried out by means of data packets on the basis of at least one shared protocol, e.g. Internet protocol, the speech information received by each subscriber through a receiver unit being converted into a voice signal and reproduced through an earpiece unit and the voice signal generated by each subscriber by way of a speaker unit being converted into speech information and transmitted in a transmitter unit, the speech information received through the receiver unit or the speech information converted into a voice signal being delayed and weighted with coefficients and subtracted from the voice signal generated in the speaker unit or from the voice signal converted into a corresponding speech information for echo cancellation.

It is the object of the invention to indicate such a method by which echo cancellation can be realized even with long and changing delay times within the data transmission network.

This is achieved in accordance with the invention in that the voice signal or speech information producing the echo(s) that has been delayed by at least the network delay time by means of a delay line and weighted with the coefficients is subtracted from the echo signal created at the respective other subscriber and transmitted over the data transmission network on the side of the subscriber connected to the first one.

As a result thereof, the echos which return to the subscriber that had them produced through transmission in packet oriented networks may also be erased. In taking into consideration the network delay time when delaying the originally produced voice signal(s) or speech information, the therefore required expenditure can be substantially reduced. Subtraction may be carried out either in the digital or in the analog form.

In another embodiment of the invention there may be provided that the loss of data packets arising from transmission is detected and the subtraction of the corresponding, delayed voice signal or of the corresponding, delayed speech information is suppressed accordingly.

In this way, an inverse signal is prevented from forming that would not cause an echo to be erased when a lost data fails to appear but that would give rise to an echo signal itself.

There may further be provided that, when one or several data packets have got lost, the respective preceding voice data packet is repeated.

The receiver unit repeats the last data packet in order to bridge the pause occurring when a data packet gets lost. Here, the echo cancellation unit would generate a wrong signal since it is fed with the data sent on its side. The receiver unit however hereby doubles the old data packet which do not match the originally sent data. As a result thereof, either the echo cancellation is blocked or the last data packet is fed once more at the subtraction input of the echo cancellation unit.

Accordingly, there may be provided that, on repeating the respective preceding voice data packet, the subtraction of a corresponding voice signal or of a corresponding speech information on the side of the connected subscriber is suppressed. Any echo erasure is thus prevented and inverse voice signals or inverse speech information within the echo cancellation unit cannot be produced wrongly.

According to another variant of the invention there may be provided that, on repeating the respective preceding voice data packet, the mating, stored voice signal or the mating, stored speech information respectively of the connected subscriber is subtracted with delay and weighting. As a result, the respective right inverse voice signal or the respective right inverse speech information can be subtracted for the repeated voice data packets.

There may be further provided that the coefficients k of the delay line are set to zero when the

network delay time changes.

In case no statement can be made with regard to the height of the change, this measure makes sure that no wrong echos are generated and reproduced.

According to another variant there may be provided that the change in network delay time is measured and that the values of the coefficients k assigned to the delay elements (20, 22, 23) are shifted within the delay line according to this change.

The controllability of the memory delay time can thus be realized in an easy way.

This shifting may preferably be performed automatically in that, in another embodiment of the invention, the memory delay time of the first delay element with the relatively high delay time is controlled through the voice data transmission unit by preferably interpreting the time information of the real-time protocol and wherein shifting of the coefficients k within the delay line is automatically carried out with the change of the delay time of the first delay element. Due to the relatively long delay time of the first delay element and to its controllability, echo cancellation can be accomplished with always the same computing power irrespective of the network delay time.

The invention is explained in detail herein after with the help of the exemplary embodiments illustrated in the enclosed drawing.

Fig. 1 is a block diagram of a portion of a transmission system according to the state of the art;

Fig. 2 is a block diagram of an embodiment of the transmission system according to the invention;

Fig. 3 shows an echo cancellation unit of the prior art;

Fig. 4 shows another embodiment of the transmission system of the invention;

Fig. 5 shows an echo cancellation unit for a transmission system according to the invention;

Fig. 6 is a block diagram of another embodiment of the transmission system according to the invention;

Fig. 7 is a partial view of another variant of the transmission system according to the invention and

Fig. 8 shows a detail of the block diagram according to Fig. 6.

Fig. 1 shows a portion of a transmission system for transmitting speech information for several subscribers of a data transmission network 1, a voice data connection between two subscribers 50,

51 being more specifically depicted. Each subscriber is equipped with one speaker and one earpiece unit 6, 6' and 7, 7' by way of which a voice signal can be created or made audible. The data transmission network 1, over which the voice data are transmitted, and which is indicated in Fig. 1, is the internet with the IP-(Internet Protocol)-protocol shared by all the subscribers. Within the scope of the invention, the voice data transmission may be performed over any discretional, similar data transmission network, e.g. LAN, Intranet, or the like. The data to be transmitted are divided into data packets and are interchanged between the subscribers during transmission, the routing of the data packets within the respective network being conducted according to the available transmission lines. Several interconnected networks may also be used for such a speech transmission, appropriate units must however be provided for conversion, when the protocols differ in these networks.

In many cases, transmission of the data packets takes place on the basis of the Internet Protocol (IP), each subscriber being connected to the IP-network 1 over a receiver and a transmitter unit contained in a voice data transmission unit 3.

The voice data transmission unit 3 further comprises a voice data conversion unit for converting the data packets into a voice signal and vice versa.

Furthermore different signal encoding techniques, e.g. PCM encoding, may be used that are to be taken into consideration on reception and transmission.

The speech information received through the receiving unit by subscriber 50 is converted into a voice signal in the voice data transmission unit 3 and is redirected via the output 41 toward the earpiece unit 7 where it is reproduced. The voice signal produced by subscriber 50 through the speaker unit 6 reaches via an input 42 the transmitter unit where it is converted into speech information and transmitted, said transmitter unit being contained in the voice data transmission unit 3. Once a communication between subscriber 50 and subscriber 51 has been established over the data transmission network 1, subscriber 51 can make audible the voice signal produced in the speaker unit 6 by means of his earpiece unit 7'after said voice signal has been converted into data packets and said packets transmitted and changed back. On his side, subscriber 51 can generate through the speaker unit 6' a voice signal that is audible to subscriber 50 after adequate transmission. During the conversation, the voice signal transmitted from subscriber 50 to subscriber 51 produces a reflection since said voice signal reaches the speaker unit 6' in parts through acoustic coupling or directly from the earpiece unit 7', where it is converted into a voice signal that corresponds to the transmitted voice signal with a lower amplitude.

If subscriber 50 is active and talks into the speaker unit, e.g. into a microphone 6, the acoustic coupling of the speaker unit 6 and of the earpiece unit 7', e.g. a loudspeaker, causes an echo to arise, said echo being erased by the echo cancellation unit 5' in that the delayed voice signal proceeding from the receiver unit of unit 3' to the subtraction input 11' is subtracted from the voice signal generated in the speaker unit 6'. In the same way, the acoustic signal occasioned by the earpiece unit

7 will partially reach the speaker unit 6 and would be sent back as an echo to subscriber 51 over said speaker unit. The voice signal emitted through the earpiece unit 7 is now subtracted from the signal generated in the speaker unit 6 in the echo cancellation unit 5 and is thus freed from the echo.

In case of a breakdown of the echo cancellation unit on the side of subscriber 51 or in case it has not been implemented owing to the cost it would involve, the transmission system according to Fig. 2 is extended on the side of subscriber 50.

This is achieved according to the invention by providing an additional echo cancellation unit 9 with an addition and a subtraction input 14, 15, the output 16 thereof being connected to the earpiece unit 7. The subtraction input 15 is thereby connected to the transmitter input 42 of the voice data transmission unit 3 and the addition input 14 is connected to the receiver input 41 of the voice data transmission unit 3.

The voice signal producing the echo that has been delayed by at least the network delay time by means of a delay line and weighted with the coefficients k is subtracted from the receiver signal created at the other subscriber 51 and transmitted over the data transmission network (1) on the side of the subscriber 50 connected to the first one. The transmission system according to the invention has been represented in Fig. 2 in a simplified form, analog voice signals being subtracted in the echo cancellation units 5, 9 in order to erase the echo signals. In modern systems, this echo cancellation is mainly carried out in the digital form, as shown in Fig. 6. In a general representation, an analog-to-digital converter 61 and 61' respectively and a digital-to-analog converter 60 and 60' respectively is inserted each in the path leading to the speaker units 6 and 6' and to the earpiece units 7 and 7' respectively through which the analog voice signal to be transmitted is converted into digital speech information and the digital speech information that has either been received or that proceeds from the echo cancellation unit 9 is converted into an analog voice signal. It has thus clearly been defined that the echo cancellation itself occurs in digital form in the respective echo cancellation unit 5, 9. Fig. 8. displays a detail of the block diagram according to Fig. 6 for purposes of clarity. The echo cancellation unit 9 is realized in the form of a DSP or of a fast computer. In a practical realization, the echo cancellation unit 9 can be combined with the echo cancellation unit 9 in one single computer or DSP.

While the echo cancellation unit 5 is still ... by a conventional system for suppressing echos with a delay time of 64 ms maximum, the other echo cancellation unit 9 must be capable of suppressing echo delay times ranging for example from 500 ms to 1500 ms since the full delay time of the data network 1 becomes effective between the subtraction input 15 and the addition input 14. For this reason, the known echo cancellation units are suited to be used in packet oriented transmission under very restricted conditions only. The way of operation of a known echo cancellation unit is therefore explained with the help of the representation in Fig. 3, the diagrams a, b, c, d illustrating the time taken by the respective signal at the places marked with the arrows pointing away from the diagrams. Again it is assumed that the echo cancellation can be performed in both the analog and the digital form. When the echo cancellation unit is of a digital design, the signals shown are digitized samplings

or speech information.

The voice signal proceeding from the receiver unit of the voice data transmission unit 3 is depicted by way of example in the diagram a of Fig. 3 and generates at the subscriber's 50 place one or several echos depending upon the type and length of the path on which the sound travels between the earpiece unit 7 and the speaker unit 6. As compared to diagram a, in diagram b, the echo signal is represented as an attenuated signal that is time-delayed accordingly. This signal b reaches the addition input 12 of the echo cancellation unit 9. Signal a is fed at the subtraction input 11 of the echo cancellation unit 9 so that it is inverted and delayed in a delay line with delay elements 21, 22, 23. Tappings 31, 32 at which the delayed signal can be tapped, attenuated with a coefficient k and brought to the summing point designated at "+" are located between said delay elements. Even multiple echos can therefore be simultaneously compensated by way of these different tapping locations since an appropriate tapping location can be provided for each possible time position of the echo signal.

The substantially echo-free signal d is generated by subtracting the delayed and attenuated signal a from the echo signal produced in the speaker unit 6. Several known methods are suited to compute the coefficients, a (N)LMS (Normalized) Least Mean Square-Fit can for example be performed by which the coefficients may be calculated for a minimum of the compensated signal.

Such an echo cancellation unit of the prior art according to Fig. 3 is very well suited for conventional telephony applications but it performs its functions in a restricted manner only when speech transmission is to be carried out over a packet oriented data network. Both the complexity of echo cancellation and the adjusting time of the coefficients k increase proportionally to the maximum echo delay time on account of the high delay time within the data network 1.

According to an embodiment of the invention, this problem can be eliminated by providing the first delay element 20 of the delay line with a relatively high memory delay time as this is shown in Fig. 5.

Instead of employing a customary delay line provided with tappings from the time 0 up to the maximum echo delay time, the delay line starts with a delay element 20 with a relatively long memory delay time that substantially corresponds to the minimum overall delay time of the voice data signals in both directions of the data network. Then, the echo cancellation unit 9 is not effective for echos ranging from 0 to 800 ms like it was heretofore known for example, but only for echos ranging from 600 ms to 800 ms. On account of the overall delay time through the data network 1, no echos can occur in this initial period of time. The advantage of this basic delay lies in the fact that the echo cancellation unit needs much less computing power of a signal processor for example and also much less memory since both less coefficients k must be calculated and less multiplications and additions respectively need to be carried out for computing the inverse delayed signal.

Echo cancellation may be further enhanced when, according to an exemplary embodiment of the

invention shown in Fig. 4, the additional echo cancellation unit 9 is provided with a control input 17 for controlling a memory delay time of the voice signal or of the speech information respectively that corresponds to the minimum delay time of the network, that is the afore mentioned overall delay time in both directions, the control input 17 being connected with the output of a control unit 18 connected to the voice data transmission unit 3. Like in the exemplary embodiment illustrated in Fig. 6, appropriate analog-to-digital converters and digital-to-analog converters respectively, and digital echo cancellation units 5, 9 can be again provided for.

As a result thereof, different conditions can be signalized to the echo cancellation unit 9 via the control unit 18. A voice data packet that has got lost can thus be detected among the incoming data for example and the echo cancellation unit 9 can be controlled accordingly so that no wrong inverted signal c (Fig. 3) is produced that would not meet any matching signal b (Fig. 3).

It is also customary to interrupt transfer during pauses occurring in speech. In order to prevent the missing data packets from causing the echo erasure times to be shifted, a recalculation of the coefficient may for example be prevented.

On repeating the preceding voice data packet, it may also be attempted to produce the corresponding inverse signal. This is only possible though, when the voice data required therefore are still contained in the delay line. For this purpose, said delay line should be realized accordingly longer.

In data networks in which the overall delay time can change, two improvements can alternatively be effected.

If the change in the delay time is known but cannot be quantified, the coefficients k of the echo cancellation unit are simply set to zero in order to prevent a double echo from becoming audible. But if the change in the network delay time is known by way of an appropriate device, e.g. by interpreting the time information of the real-time protocol, a permanent tracking of the delay time of the first delay element 20 (Fig. 5) with a relatively long memory delay time may occur so that very efficient erasure with little computing and storing capacity may be carried out.

The change in the network delay time is thereby measured and the values of the coefficients k assigned to the delay elements 20, 22, 23 within the delay line are shifted back and forth with respect to time in accordance with this change. The memory delay time of the first delay element 20 with the relatively high delay time can be controlled by way of the voice data transmission unit 3 and the shifting of the coefficients k within the delay line can be performed automatically with the change of the delay time of the first delay element 20.

The exemplary embodiment according to Fig. 7 shows a variant of the invention in which the speaker unit and the earpiece unit are realized by a subscriber terminal 70 that is connected over a subscriber's line 71 to a user interface 72 that may be provided in the analog or digital form. The echo

cancellation units 5, 9 of Fig. 6 are combined into an echo cancellation unit 73. The further connection to the data transmission network occurs in the same way as in Fig. 6. As a result thereof, the echo cancellation according to the invention can also be used when a user interface is incorporated.

CLAIMS

1. Transmission system for transmitting speech information within at least one data transmission network, such as LAN, Intranet, Internet, connecting several subscribers, in which transmission is carried out by means of data packets on the basis of at least one shared protocol, e.g. Internet protocol, and each subscriber is connected to the network via a voice data transmission unit (3), possibly an analog-to-digital converter (61) and a digital-to-analog converter (60), comprising a transceiver unit and a voice data conversion unit as well as a speaker and an earpiece unit (6, 7) connected to said voice data transmission unit, the speaker unit (6) or the inserted analog-to-digital converter (61) being connected to the transmission unit via an addition input (12) of an echo cancellation unit (5) and the output (13) thereof and a subtraction input (11) of the echo cancellation unit (5) being connected to the earpiece unit (7)or to the inserted digital-to-analog converter (60), **wherein** an additional echo cancellation unit (9) with an addition and a subtraction input (14, 15) is provided, the output (16) thereof being connected to the earpiece unit (7) or to the inserted digital-to-analog converter (60) and wherein the subtraction input (15) is connected to the transmitter unit of the voice data transmission unit (3) and the addition input (14) is connected to the receiver unit of the voice data transmission unit (3).

2. Transmission system according to claim 1, **wherein** the additional echo cancellation unit (9) is provided with a control input (17) for controlling a memory delay time of the voice signal or of the speech information that corresponds to the minimum delay time of the network, said control input (17) being connected to the output of a control unit (18) connected to the voice data transmission unit (3).

3. Transmission system with an echo cancellation unit comprising a delay line consisting of several delay elements, **wherein** the first delay element (20) of the delay line has a relatively high memory delay time that is substantially equal to the minimum overall propagation time of the voice data signals in both directions of the data network (1).

4. Transmission system according to claim 3, **wherein** the memory delay time of the first delay element (20) of the delay line may be controlled by way of the voice data transmission unit (3), preferably by interpreting the time information of the real-time protocol.

5. Method of transmitting speech information within at least one data transmission network, such as LAN, Intranet, Internet, connecting several subscribers, in which transmission is carried out by means of data packets on the basis of at least one shared protocol, e.g. Internet protocol, the speech information received by each subscriber through a receiver unit being converted into a voice signal and reproduced through an earpiece unit (7) and the voice signal generated by each subscriber through a speaker unit (6) being converted into speech information and transmitted in a transmitter

unit, the speech information received through the receiver unit or the speech information converted into a voice signal being delayed and weighted with coefficients and subtracted from the voice signal generated in the speaker unit (6) or from the voice signal converted into a corresponding speech information for echo cancellation, **wherein** the voice signal or speech information producing the echo(s) that has been delayed by at least the network delay time by means of a delay line (21, 22, 23) and weighted with the coefficients (k) is subtracted from the echo signal created at the respective other subscriber (50, 51) and returned over the data transmission network (1) on the side of the subscriber (50, 51) connected to the first one.

6. Method according to claim 5, **wherein** the loss of data packets arising from transmission is detected and the subtraction of the corresponding, delayed voice signal or of the corresponding, delayed speech information is suppressed accordingly.

7. Method according to one of the claims 4 or 5, **wherein**, when one or several data packets have got lost, the respective preceding voice data packet is repeated.

8. Method according to claim 7, **wherein,** on repeating the respective preceding voice data packet, the subtraction of a corresponding voice signal or of a corresponding speech information on the side of the connected subscriber is suppressed.

9. Method according to claim 7, **wherein,** on repeating the respective preceding voice data packet, the mating, stored voice signal or the mating, stored speech information respectively of the connected subscriber is subtracted with delay and weighting.

10. Method according to claim 5, **wherein** the coefficients (k) of the delay line (20, 22, 23) are set to zero when the network delay time changes.

11. Method according to claim 5, **wherein** the change in network delay time is measured and the values of the coefficients k assigned to the delay elements (20, 22, 23) are relocated within the delay line according to this change.

12. Method according to claim 11, **wherein** the memory delay time of the first delay element (20) with the relatively high delay time is controlled through the voice data transmission unit (3) by preferably interpreting the time information of the real-time protocol and wherein relocation of the coefficients k within the delay line is automatically carried out with the change of the delay time of the first delay element   (20).
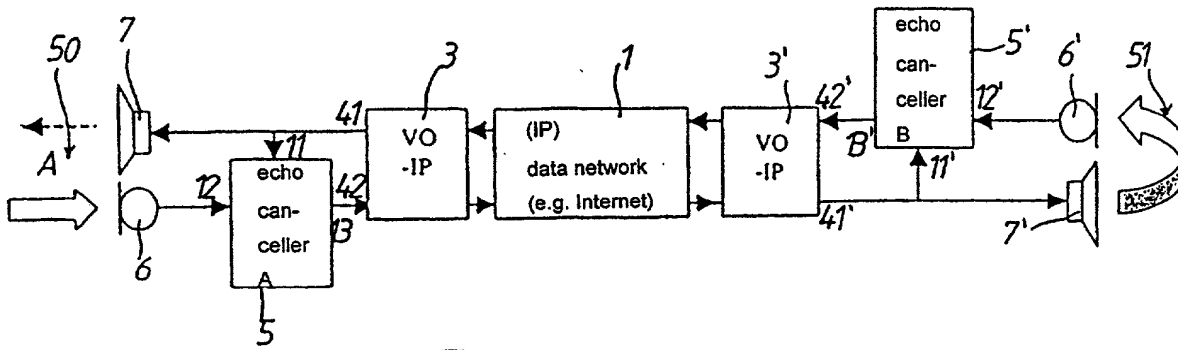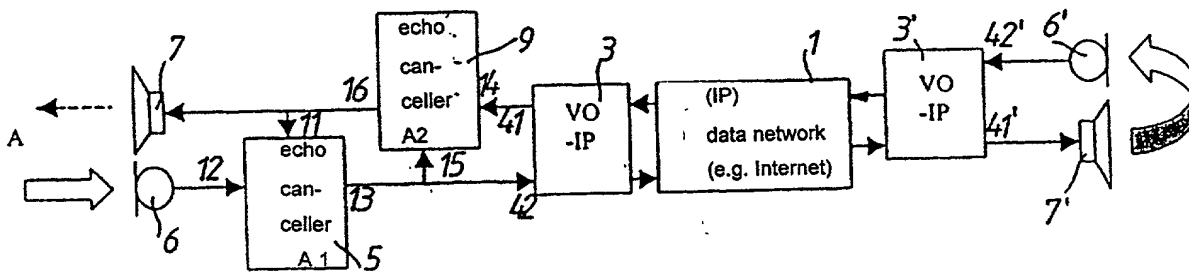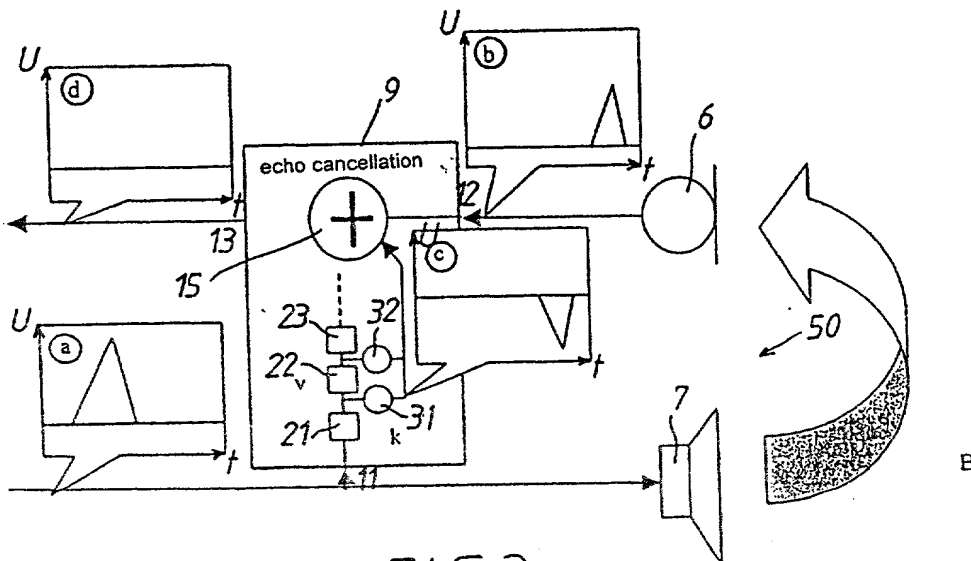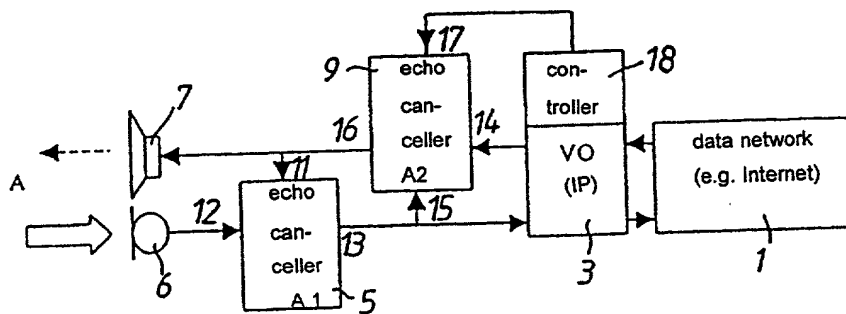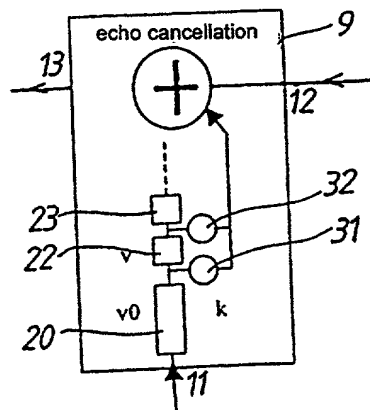
1/3

FIG.1

FIG.2

FIG.3

2/3

FIG.4

FIG.5
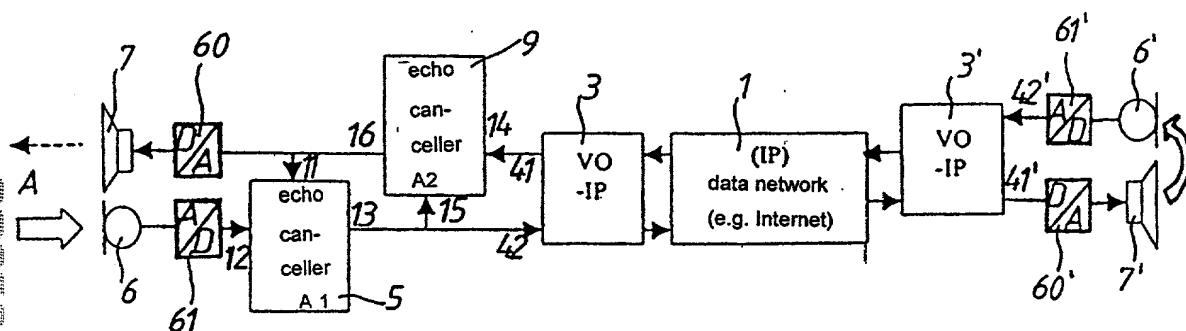
FIG.6
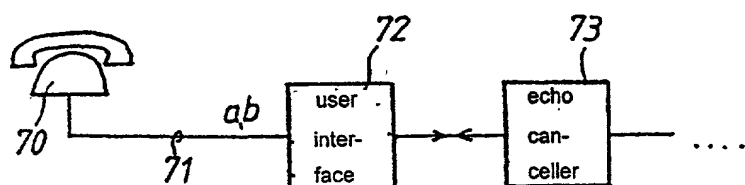

FIG.7

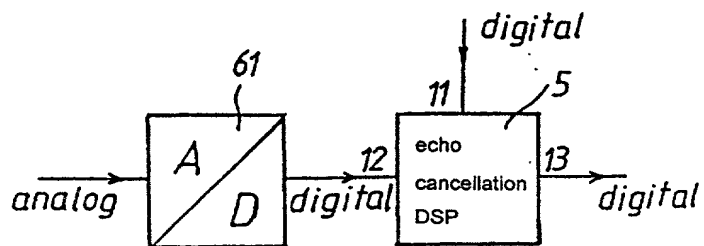

FIG.8

COMBINED DECLARATION FOR PATENT APPLICATION AND POWER OF ATTORNEY (Includes Reference to Provisional and PCT International Applications)	Attorney's Docket No. 032287-023

As a below named inventor, I hereby declare that:
My residence, post office address and citizenship are as stated below next to my name;
I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

TRANSMISSION SYSTEM WITH ECHO CANCELLATION

the specification of which (check only one item below):

☐ is attached hereto.

☐ was filed as United States application

Number _____

on _____

and was amended

on _____ (if applicable).

☒ was filed as PCT international application

Number _____ PCT/AT99/00278 _____

on _____ 17 November 1999 _____

and was amended

on _____ (if applicable).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose to the Office all information known to me to be material to patentability as defined in Title 37, Code of Federal Regulations, §1.56.

I hereby claim foreign priority benefits under Title 35, United States Code, §119 (a)-(e) of any foreign application(s) for patent or inventor's certificate or of any PCT international application(s) designating at least one country other than the United States of America listed below and have also identified below any foreign application(s) for patent or inventor's certificate or any PCT international application(s) designating at least one country other than the United States of America filed by me on the same subject matter having a filing date before that of the application(s) of which priority is claimed:

PRIOR FOREIGN/PCT APPLICATION(S) AND ANY PRIORITY CLAIMS UNDER 35 U.S.C. §119:

COUNTRY (if PCT, indicate "PCT")	APPLICATION NUMBER	DATE OF FILING (day, month, year)	PRIORITY CLAIMED UNDER 35 U.S.C. §119
AUSTRIA	A 2189/98	30 December 1998	_X_ Yes _ No
			_ Yes _ No
			_ Yes _ No
			_ Yes _ No
			_ Yes _ No

(05/01)

I hereby claim the benefit under Title 35, United States Code § 119(e) of any United States provisional application(s) listed below.

_____   _____
(Application Number)   (Filing Date)

_____   _____
(Application Number)   (Filing Date)

I hereby claim the benefit under Title 35, United States Code, §120 of any United States applications(s) or PCT international application(s) designating the United States of America that is/are listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in that/those prior application(s) in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose to the Office all information known to me to be material to the patentability as defined in Title 37, Code of Federal Regulations §1.56, which became available between the filing date of the prior application(s) and the national or PCT international filing date of this application:

PRIOR U.S. APPLICATIONS OR PCT INTERNATIONAL APPLICATIONS DESIGNATING THE U.S. FOR BENEFIT UNDER 35 U.S.C. §120:

U.S. APPLICATIONS		STATUS (check one)		
U.S. APPLICATION NUMBER	U.S. FILING DATE	PATENTED	PENDING	ABANDONED
PCT APPLICATIONS DESIGNATING THE U.S.				
PCT APPLICATION NO.	PCT FILING DATE	U.S. APPLICATION NUMBERS ASSIGNED (if any)		

(05/01)

I hereby appoint the following attorneys and agent(s) to prosecute said application and to transact all business in the Patent and Trademark Office connected therewith and to file, prosecute and to transact all business in connection with international applications directed to said invention:

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

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POST OFFICE ADDRESS		
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RESIDENCE	CITIZENSHIP	
POST OFFICE ADDRESS		

(05/01)